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A MODIFIED FM SIGNAL FOR USE WITH AN ADAPTIVE ARRAY.(U)

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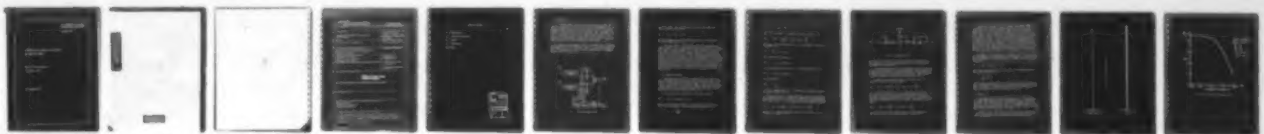

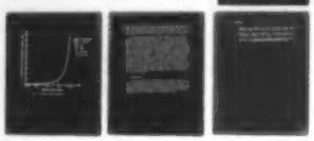

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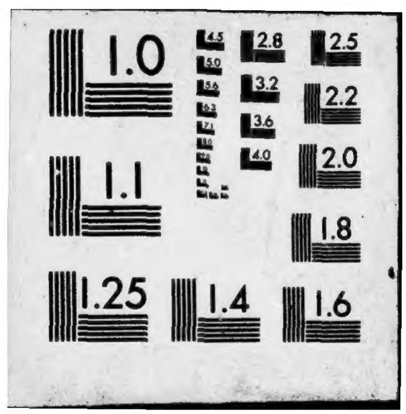
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OHIO STATE UNIVERSITY
COLUMBUS, OHIO

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20. ABSTRACT (Continue on reverse side if necessary and identify by block number)

This ~~interim~~ report discusses the use of adaptive arrays with FM signals. The objective of this work is to develop an approach for integrating an adaptive array into a conventional FM communication system. A method of modifying an FM signal to allow the array to distinguish between it and interference is currently under study.

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I. INTRODUCTION

This interim report discusses the use of adaptive arrays with FM signals. The objective of this work is to develop an approach for integrating an adaptive array into a conventional FM communication system. A method of modifying an FM signal to allow the array to distinguish between it and interference is currently under study. The method consists of including an extra digital pseudonoise code modulation to the signal in addition to the FM. This interim report contains a brief description of the modified signals under study and presents a few preliminary results on the array behavior with such signals. A more complete report on this subject is planned for the future.

An adpative array based on the LMS algorithm[1,2] has the general structure shown (for two elements) in Fig. 1. The incoming signal from each element, $y_i(t)$, is split into in-phase and quadrature components $x_i(t)$. Each component is multiplied by a weight w_i and then summed to produce the array output s(t),

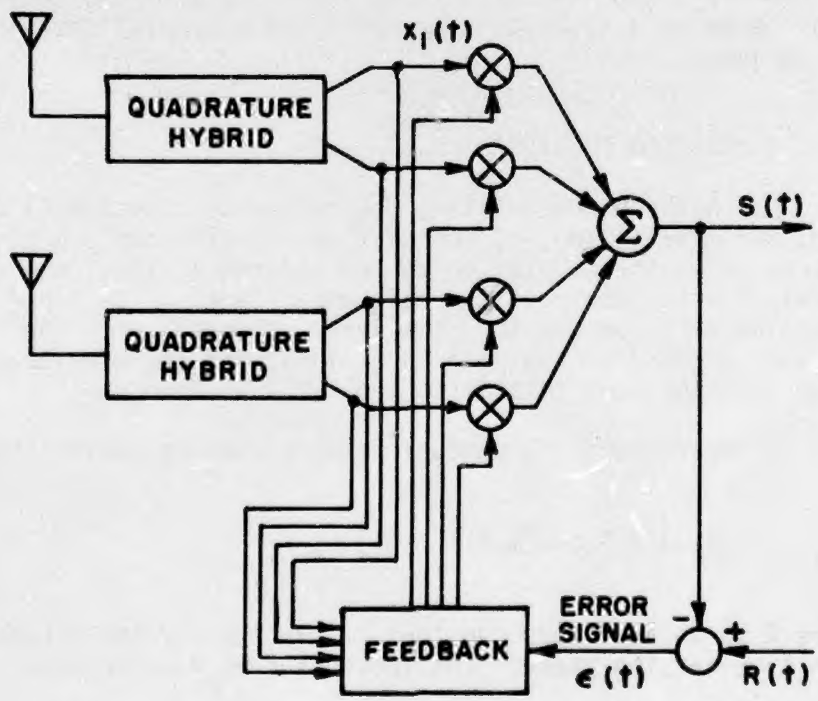

Fig. 1. The LMS adaptive array.

(1) $$s(t) = \sum_{i=1}^{4} w_i \, x_i(t) \quad .$$

The error signal $\varepsilon(t)$ is obtained by subtracting the output of the array from a reference signal $R(t)$,

(2) $$\varepsilon(t) = R(t) - s(t) \quad .$$

The array feedback based on the LMS algorithm adjusts the weights w_i to minimize the mean-square value of $\varepsilon(t)$.

The signal used for the reference signal $R(t)$ determines which received signals will be accepted by the array and which will be rejected. When a signal received by the array is highly correlated with the reference signal, the array feedback retains that signal in the output. A signal uncorrelated with the reference signal is nulled by the array. Hence the array can be used to protect a communication system from interference if the reference signal can be made strongly correlated with the desired signal but uncorrelated with the interference. Such a reference signal is usually obtained by processing the array output in some manner that preserves the desired signal but destroys the correlation in the interference components. For this to be possible, it is necessary that the desired signal differ in some known way from the interference. We describe below a technique for modifying a conventional FM signal so this may be done.

II. A MODIFIED FM SIGNAL

One method decorrelating the reference signal $R(t)$ from interference, but maintaining its correlation with the desired signal, is to add biphase switching modulation to the reference signal and the desired signal. If the phase of the reference signal is switched back and forth π radians more rapidly than the array feedback loops can track, the time-averaged produce of the reference signal and an interference signal, which does not have this switching, will be zero.

A conventional FM communication signal may be written

(3) $$s(t) = A \cos[\omega_c t + \theta(t)]$$

where A is an amplitude constant, ω_c is the carrier frequency and $\theta(t)$ is a time-varying phase. The instantaneous frequency, ω_i, is

(4) $$\omega_i = \omega_c + \frac{d\theta(t)}{dt}$$

In ordinary FM, ω_i is linearly related to a modulating signal f(t),

(5) $$\omega_i = \omega_c + K\,f(t)$$

where K is a constant. The signal s(t) then has the form

(6) $$s(t) = A\cos\left[\omega_c t + K\int_0^t f(t')dt' + \theta_0\right]$$

where θ_0 is the initial pahse at t=0. If, for example, f(t) is sinusoidal at frequency ω_m,

(7) $$f(t) = a\cos\omega_m t$$

the instantaneous freqiency ω_i is

(8) $$\omega_i = \omega_c + \Delta\omega\cos\omega_m t.$$

$\Delta\omega$ is called the deviation ($\Delta\omega$ = Ka). The phase variation $\theta(t)$ is

(9) $$\theta(t) = \frac{\Delta\omega}{\omega_m}\sin\omega_m t + \theta_0$$

and the quantity β,

(10) $$\beta = \frac{\Delta\omega}{\omega_m}$$

is called the modulation index. In general, the bandwidth occupied by the signal increases with β.

We are interested in receiving desired signals of the above type with the adaptive array. To obtain the reference signal-interference signal decorrelation discussed above, we add an extra phase modulation $\phi(t)$ to this signal. I.e., we suppose the desired signal has the form

(11) $$s(t) = A\cos\left[\omega_c t + \int_0^t f(t')dt' + \theta_0 + \phi(t)\right]$$

where $\phi(t)$ is a digital waveform with the values 0 and π, on intervals of length T, as shown in Fig. 2.

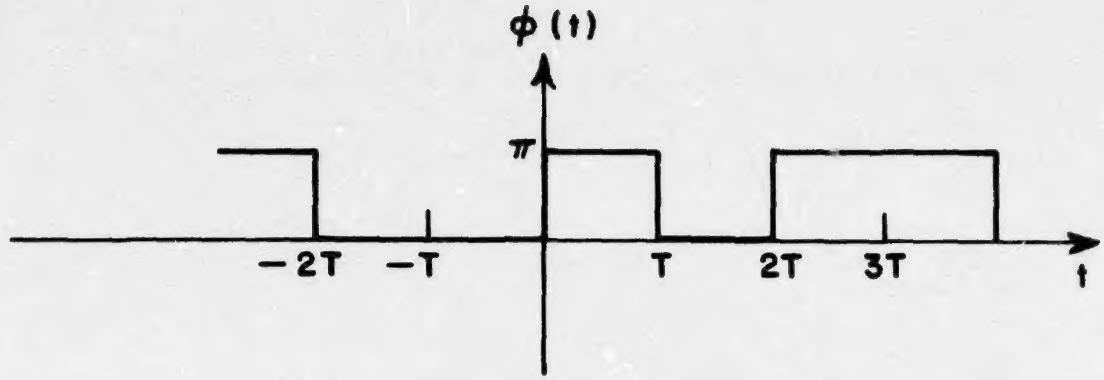

Fig. 2. The waveform $\phi(t)$.

This modulating waveform $\phi(t)$ would be added to the desired signal at the transmitter. We obtain the waveform $\phi(t)$ from a maximal length pseudonoise code generator[3].

With this modulation present on the desired signal, the same modulation can be introduced on the reference signal. The presence of $\phi(t)$ on the reference signal will make the time-average produce between the reference signal and an interference signal be zero. However, the desired signal remains correlated with the reference signal.

Ideally, when the desired signal has the form in Eq. (11), the reference signal should also be given by

$$R(t) = A \cos\left[\omega_c t + \int_0^t f(t')dt' + \theta_0 + \phi(t)\right] \quad , \tag{12}$$

i.e., it should be identical to s(t). The problem, of course, is that f(t) and $\phi(t)$ are not known at the receiving site ahead of time. (If they were, there would be no need for the antenna!) Instead, it is necessary to obtain estimates of f(t) and $\phi(t)$ (which we denote by $\hat{f}(t)$ and $\hat{\phi}(t)$) by demodulating the received signal. From these, a reference signal

$$R(t) = A \cos\left[\omega_c t + \int_0^t \hat{f}(t')dt' + \hat{\phi}(t)\right] \tag{13}$$

may be constructed. This reference signal will be suitable only if $\hat{f}(t)$ and $\hat{\phi}(t)$ are sufficiently good estimates of f(t) and $\phi(t)$.

To make use of this technique, it must be possible to demodulate both $f(t)$ and $\phi(t)$ separately, i.e., to extract each waveform without interference from the other. Feasible methods of doing this appear to be available and are currently being studied; however, the results are incomplete at this time, and these techniques will be discussed in a future report. It will suffice here to note that since $\phi(t)$ is assumed to be a maximal length PN sequence, the waveform for $\phi(t)$ is known ahead of time at the receiver. The only requirement in demodulating $\phi(t)$ is to determine its timing. Also, several methods of demodulating $f(t)$ are available, but these techniques result in a time delay between $\hat{f}(t)$ and $f(t)$ (due to filtering required). To maintain correlation between the reference signal and the desired signal this time delay must be minimized. On the other hand, minimizing the time delay results in a noisier estimate $\hat{f}(t)$. These tradeoffs and their effects on array performance are being studied and the results will be discussed in a future report.

In addition to using the type of reference signal given in Eq. (13), another approach is also possible and is being studied. The problem of time delay between $\hat{f}(t)$ and $f(t)$ may be eliminated by simply not including the FM modulation in the reference signal. Instead, a reference signal of the form

$$R(t) = A\cos[\omega_c t + \hat{\phi}(t)] \tag{14}$$

may be used. This approach is suitable only if the frequency deviation due to the modulation term

$$\int_0^t f(t')dt'$$

in the desired signal is small, i.e., only for low values of modulation index. However, this reference signal is much easier to generate, since $\hat{f}(t)$ is not needed. Simulations of adaptive array performance with both types of reference signals are presently being done.

III. RESULTS

A few typical results are shown in Figs. 3-6. These curves are for a two-element array with half wavelength spacing. A desired signal is incident from broadside and a CW interference signal from 60° off broadside. The reference signal contains only the code modulation as in Eq. (14) and it is perfectly timed. The interference power is 20 dB above that of the desired signal. The desired signal contains sinusoidal FM at 4 kHz, and the PN code rate is 20 kHz.

Figures 3 and 4 show the error signal $\varepsilon(t)$ and the array output signal for this case. It is seen that the interference is nulled during the initial weight transient and that the desired signal appears (the

Fig. 3. The error signal $\varepsilon(t)$.

Fig. 4. The array output signal.

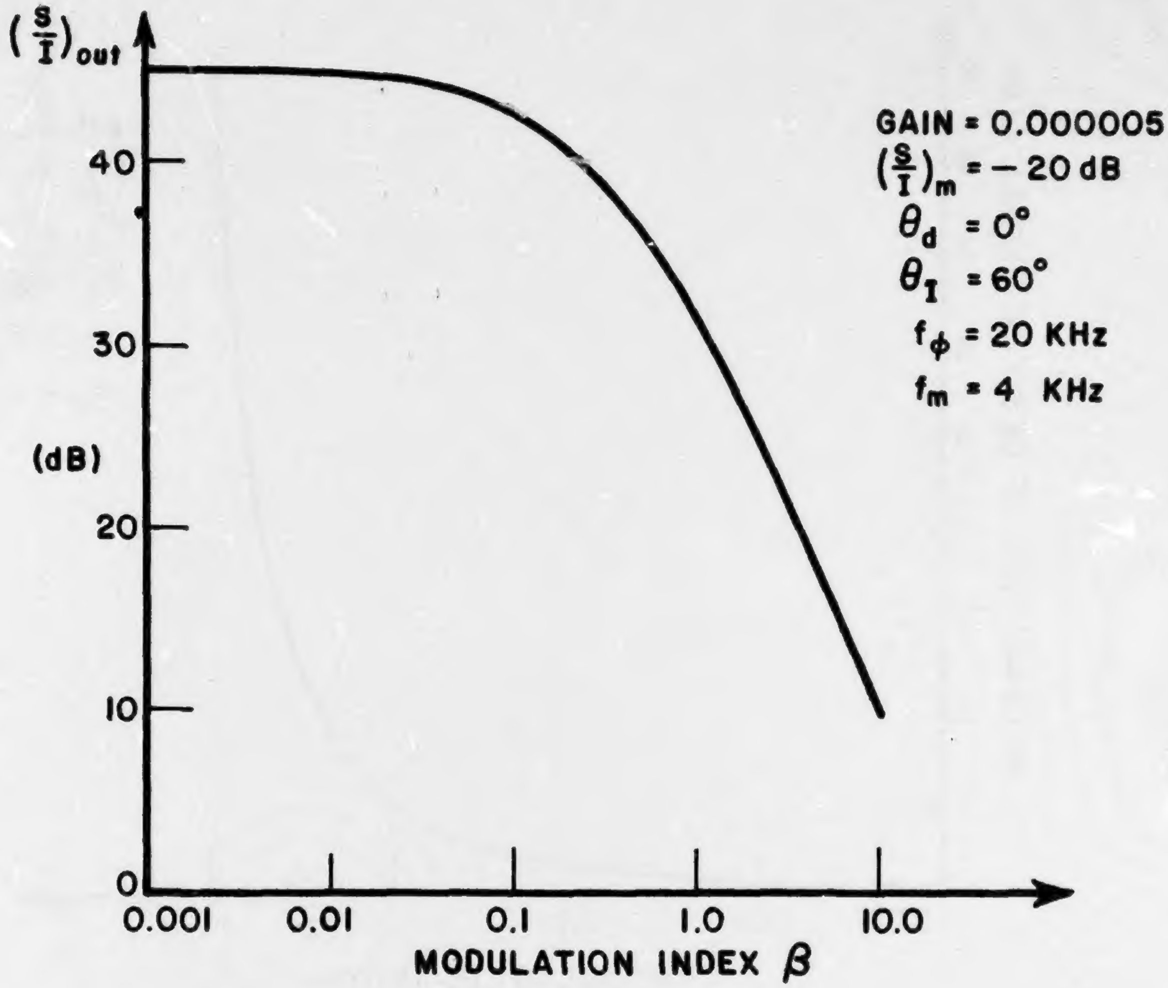

Fig. 5. Output signal-to-interference ratio.

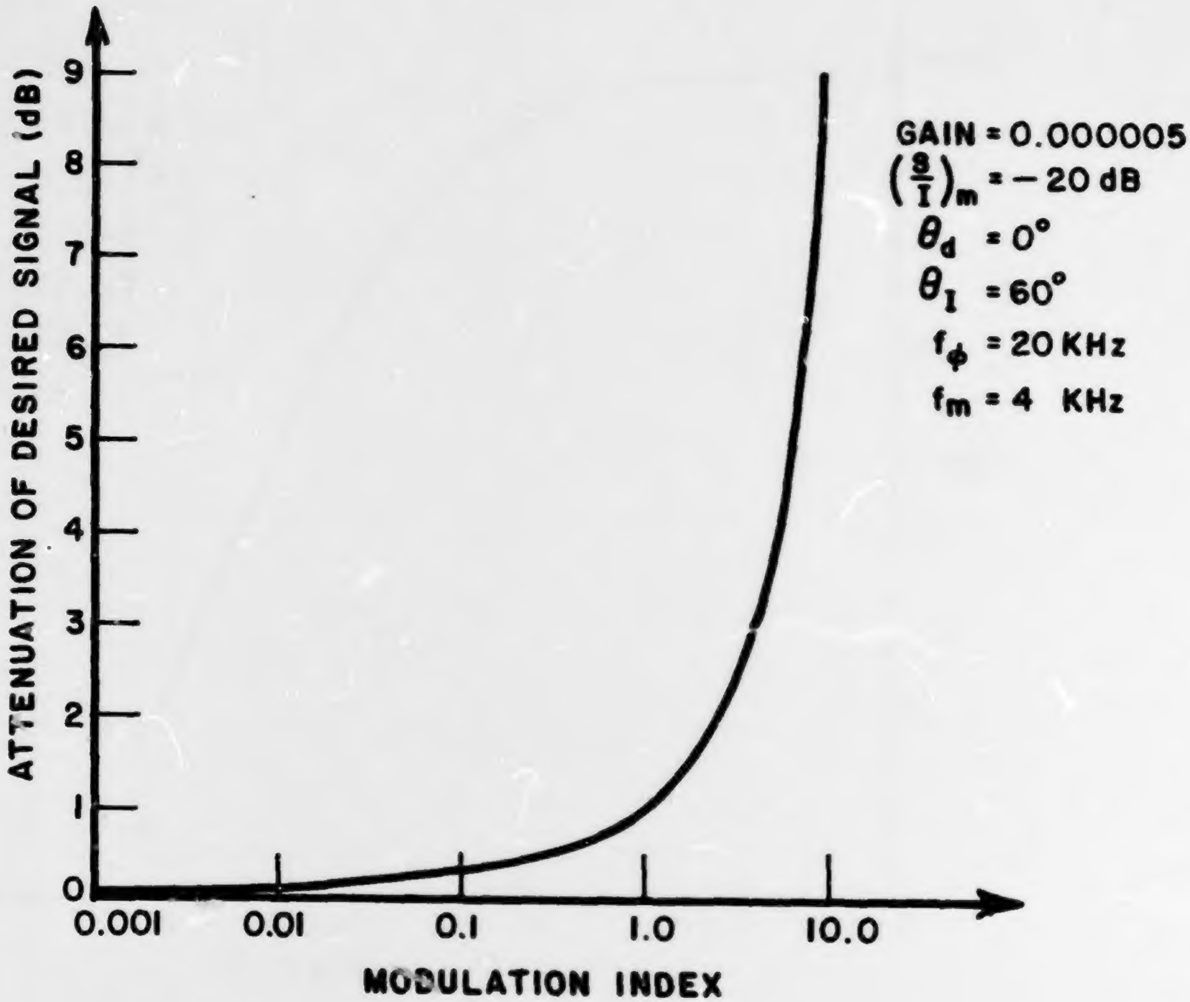

Fig. 6. Desired signal attenuation.

phase switching may be seen) after the interference has been removed. Figure 5 shows the output signal-to-interference ratio (SIR) from the array after adaptation, as a function of the modulation index β. It is clear that the output SIR begins to drop when β reaches unity. Figure 6 shows the desired signal attenuation versus β under the same conditions.

In the simulation shown above, the frequency of $\phi(t)$ was 20 kHz and the frequency modulation was at 4 kHz. In this case, $\phi(t)$ term produces a "spread spectrum" condition. In this particular case, these numbers were chosen to obtain a short simulation time on the computer. However, our ultimate objective is to have the spectrum of $s(t)$, with the $\phi(t)$ modulation, fall within the same bandwidth as it would without $\phi(t)$. I.e., we do not wish to spread the spectrum. If $\phi(t)$ can be added without altering the bandwidth, this technique can be used with existing frequency channel allocations. To add $\phi(t)$ without increasing the bandwidth requires the frequency of $\phi(t)$ to be within the bandwidth occupied by $f(t)$. More specifically, the frequency of $\phi(t)$ will have to be at the low end of the frequency range of $f(t)$, to minimize the PM-to-AM conversion that will occur at the transition times of $\phi(t)$ when the transmitted signal is bandlimited. Such low switching rates on $\phi(t)$, in turn, will require that the time constants of the LMS loops be long compared with the interval T (in Fig. 2). While this condition is simple to achieve in an analog hardware adaptive array, it is difficult to simulate on a digital computer because computer running time can become excessive. For this reason, higher frequencies for $\phi(t)$ are often used for simulation purposes.

IV. CONCLUSIONS

A method of modifying a conventional FM signal to allow it to be used with an adaptive array has been briefly discussed. The method consists of adding digital biphase modulation to the desired signal. The reference signal required in the LMS adaptive array can be of two types, depending on whether an estimate of the desired signal FM modulation is included. When the FM modulation is not included, the array operates properly only for small values of the modulation index.

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